Web Real-Time Communications Deployment Guide

Product Overview

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Genesys WebRTC Service provides web browsers with Real-Time Communications (RTC) capabilities via simple JavaScript APIs. This means you can write rich, real-time multimedia applications—such as video chat—on the web, without requiring plug-ins, downloads, or installs. Genesys WebRTC Service is able to do this by using the Web Real-Time Communications technologies that have been developed by the World Wide Web Consortium and the Internet Engineering Task Force.

The key component of Genesys WebRTC Service is the Genesys WebRTC Gateway. The Gateway converts WebRTC communications to SIP, allowing for the seamless integration of WebRTC-capable devices into a SIP-based infrastructure—in particular, an architecture that uses Genesys SIP Server. From that perspective, the WebRTC Gateway plays the role of a Media Gateway / Session Border Controller.

All calls that have been initiated or received by a browser, including browser-to-browser calls, are bridged through the Gateway. This is done to ensure that every call is handled by SIP Server as a SIP call, so that all of the core Genesys features—including routing, treatments, and IVR—can be provided by SIP Server.

For more information about the Genesys WebRTC Service, see the Architecture page in the WebRTC Developer's Guide.